CLAIMS

1. A method of encoding unvoiced segments of speech, comprising:

partitioning a residual signal frame into a plurality of sub-frames;

creating a group of sub-frame gains by computing a codebook gain for each of the plurality of sub-frames;

partitioning the group of sub-frame gains into sub-groups of sub-frame gains;

normalizing the sub-groups of sub-frame gains to produce a plurality of normalization factors wherein each of the plurality of normalization factors is associated with one of the normalized sub-groups of sub-frame gains;

converting each of the plurality of normalization factors into an exponential form and quantizing the converted plurality of normalization factors;

quantizing the normalized sub-groups of sub-frame gains to produce a plurality of quantized codebook gains wherein each of the codebook gains is associated with a codebook gain index for one of the plurality of sub-groups;

generating a random noise signal comprising random numbers for each of the plurality of sub-frames;

selecting a pre-determined percentage of the highest-amplitude random numbers of the random noise signal for each of the plurality of sub-frames;

scaling the selected highest-amplitude random numbers by the quantized codebook gains for each sub-frame to produce a scaled random noise signal;

band-pass filtering and shaping the scaled random noise signal;

analyzing the energy of the residue signal frame and the energy of the scaled random signal to produce an energy analysis;

selecting a second filter based on the energy analysis and further shaping the scaled random noise signal with the selected filter; and

generating a second filter selection indicator to identify the selected filter.

2. The method of claim 1, wherein the partitioning a residual signal frame into a plurality of sub-frames comprises partitioning a residual signal frame into ten sub-frames.

3. The method of claim 1, wherein the partitioning the group of sub-frame gains into sub-groups comprises partitioning a group of ten sub-frame gains into two groups of five sub-frame gains each.

4. The method of claim 1, wherein the residual signal frame comprises 160 samples per frame sampled at eight kilohertz per second for 20 milliseconds. .

5. The method of claim 1, wherein the pre-determined percentage of the highest-amplitude random numbers is twenty-five percent.

6. The method of claim 1, wherein two normalization factors are produced for two sub-groups of five sub-frame codebook gains each.

7. The method of claim 1, wherein the quantizing the of sub-frame gains is performed using multi-stage vector quantization.

8. A method of encoding unvoiced segments of speech, comprising:

partitioning a residual signal frame into sub-frames, each sub-frame having a codebook gain associated therewith;

quantizing the gains to produce indices;

scaling a percentage of random noise associated with each sub-frame by the indices associated with the sub-frame;

performing a first filtering of the scaled random noise;

comparing the filtered noise with the residual signal;

performing a second filtering of the random noise based on the comparison; and

generating a second filter selection indicator to identify the second filtering performed.

9. The method of claim 8, wherein the partitioning a residual signal frame into sub-frames comprises partitioning a residual signal frame into ten sub-frames.

10. The method of claim 8, wherein the residual signal frame comprises 160 samples per frame sampled at eight kilohertz per second for 20 milliseconds.

11. The method of claim 8, wherein the percentage of random noise is twenty-five percent.

12. The method of claim 8, wherein quantizing the gains to produce indices is performed using multi-stage vector quantization.

~~13~~. A speech coder for encoding unvoiced segments of speech, comprising:

means for partitioning a residual signal frame into a plurality of sub-frames;

means for creating a group of sub-frame gains by computing a codebook gain for each of the plurality of sub-frames;

means for partitioning the group of sub-frame gains into sub-groups of sub-frame gains;

means for normalizing the sub-groups of sub-frame gains to produce a plurality of normalization factors wherein each of the plurality of

normalization factors is associated with one of the normalized sub-groups of sub-frame gains;

means for converting each of the plurality of normalization factors into an exponential form and quantizing the converted plurality of normalization factors;

means for quantizing the normalized sub-groups of sub-frame gains to produce a plurality of quantized codebook gains wherein each of the codebook gains is associated with a codebook gain index for one of the plurality of sub-groups;

means for generating a random noise signal comprising random numbers for each of the plurality of sub-frames;

means for selecting a pre-determined percentage of the highest-amplitude random numbers of the random noise signal for each of the plurality of sub-frames;

means for scaling the selected highest-amplitude random numbers by the quantized codebook gains for each sub-frame to produce a scaled random noise signal;

means for band-pass filtering and shaping the scaled random noise signal;

means for analyzing the energy of the residue signal frame and the energy of the scaled random signal to produce an energy analysis;

means for selecting a second filter based on the energy analysis and further shaping the scaled random noise signal with the selected filter; and

means for generating a second filter selection indicator to identify the selected filter.

14. The speech coder of claim 13, wherein the means for partitioning a residual signal frame into a plurality of sub-frames comprises means for partitioning a residual signal frame into ten sub-frames.

15. The speech coder of claim 13, wherein the means for partitioning the group of sub-frame gains into sub-groups comprises means for partitioning a group of ten sub-frame gains into two groups of five sub-frame gains each.

16. The speech coder of claim 13, wherein the means for selecting a pre-determined percentage of the highest-amplitude random numbers comprises a means for selecting twenty-five percent of the highest-amplitude random numbers.

17. The speech coder of claim 13, wherein the means for normalizing the subgroups comprises means for producing two normalization factors for two sub-groups of five sub-frame codebook gains each.

18. The speech coder of claim 13, wherein the means for quantizing the sub-frame gains comprises means for performing multi-stage vector quantization.

19. A speech coder for encoding unvoiced segments of speech, comprising:

means for partitioning a residual signal frame into sub-frames, each sub-frame having a codebook gain associated therewith;

quantizing the gains to produce indices;

means for scaling a percentage of random noise associated with each sub-frame by the indices associated with the sub-frame;

means for performing a first filtering of the scaled random noise;

means for comparing the filtered noise with the residual signal;

means for performing a second filtering of the random noise based on the comparison; and

means for generating a second filter selection indicator to identify the second filtering performed.

20. The speech coder of claim 19, wherein the means for partitioning a residual signal frame into sub-frames comprises means for partitioning a residual signal frame into ten sub-frames.

21. The speech coder of claim 19, wherein the means for scaling a percentage of random noise comprises a means for scaling twenty-five percent of the highest-amplitude random noise.

22. The speech coder of claim 19, wherein the means for quantizing the gains to produce indices comprises means for multi-stage vector quantization.

23. A speech coder for encoding unvoiced segments of speech, comprising:

a gain computation component configured to partition a residual signal frame into a plurality of sub-frames, create a group of sub-frame gains by computing a codebook gain for each of the plurality of sub-frames, partition the group of sub-frame gains into sub-groups of sub-frame gains, normalize the sub-groups of sub-frame gains to produce a plurality of normalization factors wherein each of the plurality of normalization factors is associated with one of the normalized sub-groups of sub-frame gains, and convert each of the plurality of normalization factors into an exponential form.

a gain quantizer configured to quantize the converted plurality of normalization factors to produce a quantized normaliztion factor index, and quantize the normalized sub-groups of sub-frame gains to produce a plurality of quantized codebook gains wherein each of the codebook gains is associated with a codebook gain index for one of the plurality of sub-groups;

a random number generator configured to generate a random noise signal comprising random numbers for each of the plurality of sub-frames;

a random number selector configured to select a pre-determined percentage of the highest-amplitude random numbers of the random noise signal for each of the plurality of sub-frames;

a multiplier configured to scale the selected highest-amplitude random numbers by the quantized codebook gains for each sub-frame to produce a scaled random noise signal;

a band-pass filter for eliminating for eliminating low-end and high-end frequencies from the scaled random noise signal;

a first shaping filter for perceptual filtering of the scaled random noise signal;

an unscaled band energy analyzer configured to analyze the energy of the residue signal;

a scaled band energy analyzer configured to analyze the energy of the scaled random signal, and to produce a relational energy analysis of the energy of the residual signal compared to the energy of the scaled random signal;

a second shaping filter configured to select a second filter based on the relational energy analysis, further shape the scaled random noise signal with the selected filter, and generate a second filter selection indicator to identify the selected filter.

24. The speech coder of claim 23, wherein the band pass filter and the first shaping filters are fixed filters.

25. The speech coder of claim 23, wherein the second shaping filter is configured with two fixed shaping filters.

26. The speech coder of claim 23, wherein the second shaping filter configured to generate a second filter selection indicator to identify the selected filter is further configured to generate a two bit filter selection indicator.

27. The speech coder of claim 23, wherein the gain computation component configured to partition a residual signal frame into a plurality of sub-frames is further configured to partition a residual signal frame into ten sub-frames.

28. The speech coder of claim 23, wherein the gain computation component configured to partition the group of sub-frame gains into sub-groups is further configured to partition a group of ten sub-frame gains into two groups of five sub-frame gains each.

29. The speech coder of claim 23, wherein the random number selector configured to select a pre-determined percentage of the highest-amplitude random numbers if further configured to select twenty-five percent of the highest-amplitude random numbers.

30. The speech coder of claim 23, wherein the gain computation component configured to normalize the subgroups is further configured to produce two normalization factors for two sub-groups of five sub-frame codebook gains each.

31. The speech coder of claim 23, wherein the gain quantizer is further configured to perform multi-stage vector quantization.

32. A speech coder for encoding unvoiced segments of speech, comprising:

a gain computation component configured to partition a residual signal frame into sub-frames, each sub-frame having a codebook gain associated therewith;

a gain quantizer configured to quantize the gains to produce indices;

a random number selector and multiplier configured to scale a percentage of random noise associated with each sub-frame by the indices associated with the sub-frame;

a first perceptual filter configured to perform a first filtering of the scaled random noise;

a band energy analyzer configured to compare the filtered noise with the residual signal;

a second shaping filter configured to perform a second filtering of the random noise based on the comparison, and generate a second filter selection indicator to identify the second filtering performed.

33. The speech coder of claim 32, wherein the gain computation component configured to partition a residual signal frame into sub-frames is further configured to partition a residual signal frame into ten sub-frames.

34. The speech coder of claim 32, wherein the random noise selector and multiplier configured to scale a percentage of random noise is further configured to scale twenty-five percent of the highest-amplitude random noise.

35. The speech coder of claim 32, wherein the gain quantizer configured to quantize the gains to produce indices is further configured to perform multi-stage vector quantization.

36. The speech coder of claim 32, wherein the first perceptual filter configured to perform a first filtering of the scaled random noise is further configured to filter the scaled random noise using a fixed band pass filter and a fixed shaping filter.

37. The speech coder of claim 32, wherein the second shaping filter configured to perform a second filtering of the random noise is further configured to have two fixed filters.

38. The speech coder of claim 32, wherein the second shaping filter configured to generate a second filter selection indicator is further configured to generate a two bit filter selection indicator.

39. A method of decoding unvoiced segments of speech, comprising:

recovering a group of quantized gains using received indices for a plurality of sub-frames;

generating a random noise signal comprising random numbers for each of the plurality of sub-frames;

selecting a pre-determined percentage of the highest-amplitude random numbers of the random noise signal for each of the plurality of sub-frames;

scaling the selected highest-amplitude random numbers by the recovered gains for each sub-frame to produce a scaled random noise signal;

band-pass filtering and shaping the scaled random noise signal; and

selecting a second filter based on a received filter selection indicator and further shaping the scaled random noise signal with the selected filter.

40. The method of claim 39, further comprising further filtering the scaled random noise.

41. The method of claim 39, wherein the plurality of sub-frames comprise partitions of ten sub-frames per frame of encoded unvoiced speech.

42. The method of claim 39, wherein the plurality of sub-frames comprise partitions of sub-frame gains partitioned into sub-groups.

43. The method of claim 42, wherein the sub-groups comprise partitioning a group of ten sub-frame gains into two groups of five sub-frame gains each.

44. The method of claim 41, wherein the frame of encoded unvoiced speech comprises 160 samples per frame sampled at eight kilohertz per second for 20 milliseconds.

45. The method of claim 39, wherein the pre-determined percentage of the highest-amplitude random numbers is twenty-five percent.

46. The method of claim 43, wherein two normalization factors are recovered for two sub-groups of five sub-frame gains each.

47. The method of claim 1, wherein the recovering a group of quantized gains is performed using multi-stage vector quantization.

48. A method of decoding unvoiced segments of speech, comprising:

recovering quantized gains partitioned into sub-frame gains from received indices associated with each sub-frame;

scaling a percentage of random noise associated with each sub-frame by the indices associated with the sub-frame;

performing a first filtering of the scaled random noise;

performing a second filtering of the random noise determined by a filter selection indicator.

49. The method of claim 48, comprising further filtering the scaled random noise.

50 ~~49~~. The method of claim 48, wherein the sub-frame gains comprise partitions of ten sub-frame gains per frame of encoded unvoiced speech.

51 ~~50~~. The method of claim 49, wherein the frame of encoded unvoiced speech comprises 160 samples per frame sampled at eight kilohertz per second for 20 milliseconds.

52 ~~51~~. The method of claim 48, wherein the percentage of random noise is twenty-five percent.

53 ~~52~~. The method of claim 48, wherein the recovered quantized gains are quantized by multi-stage vector quantization.

54 ~~53~~. A decoder for decoding unvoiced segments of speech, comprising:

means for recovering a group of quantized gains using received indices for a plurality of sub-frames;

means for generating a random noise signal comprising random numbers for each of the plurality of sub-frames;

means for selecting a pre-determined percentage of the highest-amplitude random numbers of the random noise signal for each of the plurality of sub-frames;

means for scaling the selected highest-amplitude random numbers by the recovered gains for each sub-frame to produce a scaled random noise signal;

means for band-pass filtering and shaping the scaled random noise signal; and

means for selecting a second filter based on a received filter selection indicator and further shaping the scaled random noise signal with the selected filter.

57 ~~54~~. The speech coder of claim 53, comprising means for further filtering the scaled random noise.

58 ~~55~~. The speech coder of claim 53, wherein the means for selecting a pre-determined percentage of the highest-amplitude random numbers of the random noise signal further comprises means for selecting twenty five percent of the highest-amplitude random numbers.

59 ~~56~~. A decoder for decoding unvoiced segments of speech, comprising:

a gain de-quantizer configured to recover a group of quantized gains using received indices for a plurality of sub-frames;

a random number generator configured to generate a random noise signal comprising random numbers for each of the plurality of sub-frames;

a random number selector configured to select a pre-determined percentage of the highest-amplitude random numbers of the random noise signal for each of the plurality of sub-frames;

a random number selector and multiplier configured to scale the selected highest-amplitude random numbers by the recovered gains for each sub-frame to produce a scaled random noise signal;

a band-pass filter and first shaping filter to filter and shape the scaled random noise signal; and

a second shaping filter configured to select a second filter based on a received filter selection indicator and further shape the scaled random noise signal with the selected filter.

~~57~~. The speech coder of claim 56, comprising a post-filter configured to further filter the scaled random noise.

~~58~~. The speech coder of claim 56, wherein the random number selector configured to select a pre-determined percentage of the highest-amplitude random numbers of the random noise signal is further configured to select twenty five percent of the highest-amplitude random numbers.

~~58~~. A speech coder for decoding unvoiced segments of speech, comprising:

means for recovering quantized gains partitioned into sub-frame gains from received indices associated with each sub-frame;

means for scaling a percentage of random noise associated with each sub-frame by the indices associated with the sub-frame;

means for performing a first filtering of the scaled random noise;

means for performing a second filtering of the random noise determined by a filter selection indicator.

~~59~~. The speech coder of claim 58, comprising means for further filtering the scaled random noise.

62 ~~60~~. The speech coder of claim 58, wherein the means for scaling a percentage of random noise associated with each sub-frame further comprises means for scaling 25% of random noise associated with each sub-frame.

63 ~~61~~. A speech coder for decoding unvoiced segments of speech, comprising:

a gain de-quantizer configured to recover quantized gains partitioned into sub-frame gains from received indices associated with each sub-frame;

a randon number selector and multiplier configured to scale a percentage of random noise associated with each sub-frame by the indices associated with the sub-frame;

a first shaping filter configured to perform a first perceptual filtering of the scaled random noise;

a second shaping filter configured to perform a second filtering of the random noise determined by a filter selection indicator.

64 ~~62~~. The speech coder of claim 61, comprising a post-filter for further filtering the scaled random noise.

65 ~~63~~. The speech coder of claim 61, wherein the random number selector and multiplier configured to scale a percentage of random noise associated with each sub-frame further is configured to scale 25% of random noise associated with each sub-frame.